Software Tools for the Laboratory of Biomedical Signals and Algorithms

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***Abstract*—The Laboratory of Biomedical Signals and Algorithms is included as an optional subject for the last year of the Telecommunication degree (Technical University of Madrid, Spain). Students are considered to be able to understand and produce different algorithms related to different bioengineering topics.**

This article shows one software tool developed by the teachers, in order to give students an initial work material to get used to these kind of algorithms and acquire some skills in developing them. A complete MATLAB Application with a graphical interface has been performed to this purpose, with particular algorithms (Chaos theory algorithms applied to voice recognition) that should be changed and adapted to others algorithms by students.

***Index Terms*—Education, Biomedical, Bioengineering, Algorithms, Signals, Software Tools.**

I. Introduction

The last 20 years have supposed a deep change in the way technology helps when dealing with medicine and biology. Medicine has obtained unvaluable new oportunities in diagnosis and patient treatment due to the evolution of biomedical technologies (ACT, PET...) that are, at last, based on signal recognition and signal process. Biomedical signals and the way they are processed with different algorithms are the basis of all this technology surrounding what is known as Bioengineering.

To university students, bioengineering is an attractive area with a lot of possibilities with which they have no firsthand experience but a lot of curiosity, with the possibility of helping people whether with diagnosis or treatment. In response, undergraduate telecommunication courses now usually include material related to biomedicine technology. Several universities include related subjects at the undergraduate level, with experimental laboratories and colaboration agreement with different hospitals and health-care institutes. This is the case of the Technical University of Madrid: the Telecommunication Degree includes the 'Biomedical Signals and Algorithms' laboratory, with three ECTS credits, as a way to familiarise students with biomedical signal processing.

The 'Biomedical Signals and Algorithms' laboratory is an optional subject for those telecommunication students that have chosen the bioengineering branch from those different branches (computer sciences, electronics, radiocommunication...) that they are offered in telecomunication degree for the fourth and fifth year. The laboratory consists of 20 computers and some medical interfaces (EKG/ECG machine, EEG machine) to acquire signals to deal with. Two young researchers (the authors) are the managers of the laboratory, the teaching, practical explanations and student evaluation. The particular methodology reinforces theoretical concepts but also allows students to gain ability to use the skills, techniques, and modern bioengineering tools that they may use in their future workplace.

The basic laboratory software used for the different laboratory tasks is MATLAB®. This math software lets the user to handle different signals in an easy way, and process them in almost whatever way. This software will allow the students, for instance, to simulate an ATC, using given training signals, applying proper processing in the Fourier domain.

Students are supposed to carry out a biomedical simulator with this software as final evaluatory work. Getting successful implies that the student is able to deal with bioengineering signals and whatever processing related.

II. Software Basis: Chaos Theory Applied to Speech Recognition

One of the elements that the evaluation of this course takes into account is the originality of the developed idea. For that reason, one important point for the laboratory application is the selection of the initial topic for the software application. Each year, it is tried to select a new area, to enhance the curiosity of students, as they are suggested to search interesting bioengineering papers and use them as the basis of their application, that is evaluated at the end of the term. The topic selected is 'Speech Recognition using Chaos Theory', for several reasons:

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- Originality of the topic.
- Variety of Algorithms already availables.
- Introduction of an aproach that differs from those conventionals (frequency domain approach).

The chaos theory is an alternative approach for a variety of different topics: speech recognition, optimization processes, image recognition, etc. The basics of this theory are quite deeply developed in literature [1],[2]. In this paper, only a slight approach of chaos theory applied to temporal signals is given, summarising the most important definitions and theorems.

A. Attractor Theory Applied to Temporal Signals.

Certain irregular and unpredictable phenomena contains a hidden order. Actually, there are such an amount of conditioning circumstances that these phenomena seem to be unpredictable and chaotic indeed. Chaos theory tries to analyse this chaotic behaviour, from a global point of view, so it is possible to measure in a particular way this rare behaviour that is called chaos. In the case of temporal signals, the set of different values (states) that the signal may acquire is called phase space (coordinate representation of its independent unknowns).

An attractor, in global terms, is the set of space phase to which the signal evolves after a long enough period of time. Therefore, it is the representation of the 'attraction' to a particular area of the space phase that the signal suffers: The part of the phase space of the dynamical system corresponding to the typical behavior is the attracting set or attractor. The attractor of a chaotic system is called strange attractor or fractal.

Mathematicians have devised many additional ways to make quantitative statements about chaotic systems. These include: fractal dimension of the attractor, Lyapunov exponents, recurrence plots, Poincaré maps, bifurcation diagrams, transfer operator, among others. Some of them are listed below.

Takens Theorem (attractor reconstruction): Takens Theorem provides the conditions under which it is possible to reconstruct an strange attractor from the observation of one unique known coordinate of a dynamic system. The reason is clear: to determine the state of an n-dimensional system, it is enough to know n independent values. In the case of a temporal signal, that is a bit more complex, as the temporal variation is the only known value. Takens suggested that it is possible to rebuild the whole m-dimensional space with a set of m temporal values of the signal, with a delay *T* and *m,* that is known as the embedding dimension and is the superior limit of the phase space complexity for the attractor . A proper selection of *m* and *T*, allow us to obtain the original attractor.

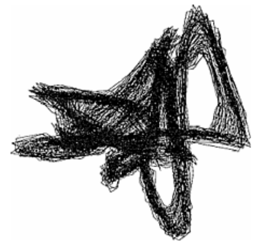


Fig. 1. Reconstructed strange attractor for /a/

Imagine $x(n)$ (n from 1 to N) as the temporal signal for which it is desired to obtain its attractor. It is necessary to configure a set of L m-dimensional vectors $V_m(i)$, with i from 1 to L, and $L=N-(m-1)T$, in order to obtain the desired attractor.

$$V_m(i) = \{x(i), x(i+T), \ldots, x(i+(m-1)T)\} \qquad (1)$$

Attractor dimension: it is the most used parameter in order to characterize an attractor and the way of defining a signal as chaotic. This parameter is based on geometric properties (table 1), which configure what is called fractal dimension. For a temporal signal, it is not possible to obtain the attractor dimension; one way to estimate it is the correlation dimension.

TABLE I
GEOMETRIC PARAMETERS

Parameter	**Formulation**
Unitary point	$(x, \Delta x)$
Statical Moment of a trayectory, from a point (M_{xD},$M_{\Delta xD}$)	$M_{x0} = \sum_{i=1}^{N} x_i - x_0$; $M_{\Delta x0} = \sum_{i=1}^{N} \Delta x_i - \Delta x_0$
Center of Mass (x_m,Δx_m)	$x_m = \frac{\sum_{i=1}^{N} x_i}{N}$; $\Delta x_m = \frac{\sum_{i=1}^{N} \Delta x_i}{N}$
Rotation radius (R_G)	$I_0 = \sum_{i=1}^{N}\left((x_i - x_m)^2 + (\Delta x_i - \Delta x_m)^2\right)$; $R_G = \sqrt{\frac{I_0}{N}}$

Correlation dimension: a way to evaluate and measure the complexity of the system, related to the number of grades of freedom. Grassberger and Procaccia [3], [4] developed a numeric computation algorithm to obtain the correlation dimension for temporal finite series. Applying Takens Theorem, $V_m(i)$ is reconstructed in terms of $x(n)$. In order to get a right estimation, some criteria have to be satisfied [5], [6].

There are other methods of measuring dimension (e.g. the Hausdorff dimension, the box-counting dimension, and the information dimension) but the correlation dimension has the advantage of being straightforwardly and quickly calculated, and is often in agreement with other calculations of dimension. The definition of the correlation integral C(ε):

$$C(\varepsilon) = \frac{1}{N^2} \sum_{k=1}^{N} \sum_{\substack{j=1 \\ j \neq k}}^{N} H(\varepsilon - \|V_m(k) - V_m(j)\|) \qquad (2)$$

Where $\|V_m(k)-V_m(j)\|$ is the euclidean norm and H is the Heaviside function:

$$H(x)=\begin{cases}0 & for \quad x \le 0\\ 1 & for \quad x>0\end{cases} \tag{3}$$

The computational costs for this algorithms are proportional to N^2. There are other algorithms that reduce significantly these computational costs: in [7] it is obtained a time consumption proportional to Nlog(N); in [8] it is obtained even better results, proportional to *N*.

Lyapunov exponents: In mathematics the Lyapunov exponent of a dynamical system is a quantity that characterizes the rate of separation (Z_0) of close trajectories. Quantitatively, two trajectories in phase space with initial separation δZ_0 diverge:

$$|\delta \mathbf{Z}(t)| \approx e^{\lambda t}|\delta \mathbf{Z}_0| \tag{4}$$

The rate of separation can be different for different orientations of initial separation vector. For this, there is a whole spectrum of Lyapunov exponents, and the number of them is equal to the number of dimensions of the *phase space*. It is useful to refer to the largest one: the Maximal Lyapunov exponent (MLE), because it determines the predictability of a dynamical system, in our case, our signal. A positive MLE is usually taken as an indication that the system is chaotic. The maximal Lyapunov exponent can be defined as follows:

$$\lambda = \lim_{t\to\infty} \frac{1}{t} \ln \frac{|\delta \mathbf{Z}(t)|}{|\delta \mathbf{Z}_0|} \tag{5}$$

B. Attractor Theory and Speech Signals.

The conventional processing techniques for speech signals are based on linear system theory, for which the frequency domain is essential. In these classical systems, all the voice cavities are considered to be a linear filter, with clear determined features. Even though all these methods are extremly useful, one new alternative is to study speech signal as non-linear system. For non-linear systems, the reconstructed phase space is equivalent to the frequency domain in a linear system. The phase space is reconstructed from a temporal string, applying Takens Theorem.

Taking again (1), where *x(t)* is the original signal, T is the delay and m the embedding dimension, it is posible to obtain the attractor which corresponds to this particular signal. Figure 2 shows an example of attractor for a speech sound.

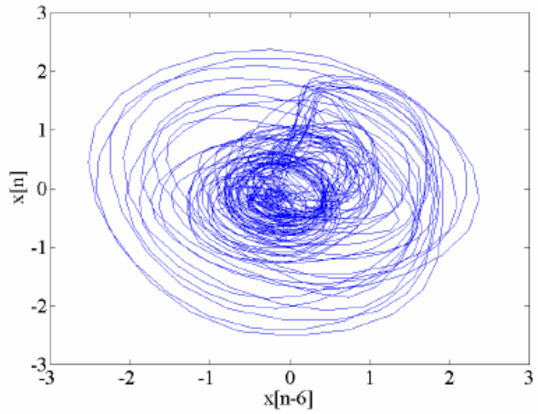


Fig. 2. Attractor for /ow/ phonem

It has been demonstrated [9] that the attractor is topologically identical to its forming signal, with the same non-linear features. Due to this aspect, attractors are essential to get features and parameters from speech signal, features that are different from the ones in a conventional spectral analysis. For instance, with the attractor correlation dimension and the attractor shape, it is possible to perform a coarse recognition algorithm, as it is easy to distinguish vowels, nasal consonants and fricative ones, only with the correlation dimension criterion, as it is shown in table 2.

- Vowels: */a/*, /e/, /i/, /o/, /u/.
- Nasals: */m/, /n/.*
- Fricatives: */f/. /s/, /z/.*

TABLE II
CORRELATION DIMENSION FOR DIFFERENT SOUNDS

Sound	*/a/*	*/e/*	*/i/*	*/o/*	*/u/*	*/m/*	*/n/*	*/f/*	*/s/*	*/z/*
D_f	*2.2*	*2.2*	*2.2*	*2.1*	*2.1*	*1.8*	*1.8*	*3.3*	*4.9*	*4.1*

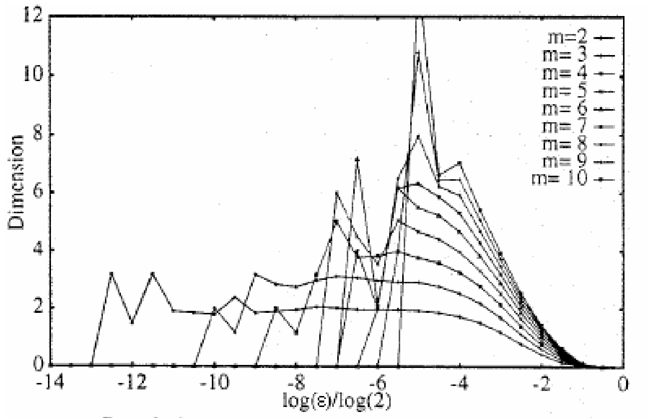
Consonant Recognition

One of the most promising applications for this attractor dimension algorithm implies the possibility of of recognition and discrimination for different consonant phonemes. Takens Theorem allow us to determine the time slot of a complete speech that corresponds to consonants.

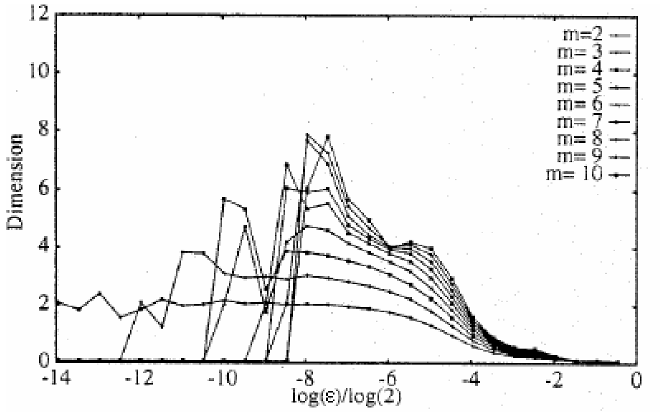
Recent trials have analysed and processed 22 consonant phonemes (telephone sound quality). Applying the Grassberger algorithm, the values for the correlation dimension for different *m* values are obtained. The correlation dimension could be plotted considering *ln(C(ε))* versus *ln(ε)* for variation *(ε)* values small enough.

$$C(\varepsilon)=\frac{1}{N^2}\sum_{k=1}^{N}\sum_{\substack{j=1\\ j\neq k}}^{N} H(\varepsilon-\|V_m(k)-V_m(j)\|) \tag{6}$$

Some significative results could be obtained from [10], as figure o and f show.



Correlation dimension of excitations for fricative /f/.



Correlation dimension of excitations for fricative /v/.

Although a complete classification method has not been implemented, it is possible to distinguish different consonants with this kind of graphical results. The correlation dimension for diverse scales and *m* values is an essential source to determine non-linear features: every consonant phoneme has its own different local dimensions. Table 4 presents a set of correlation dimension values for an amount of phonemes [10].

Pho-nem	Type, Voiced/ Unvoiced	*m*=8	*m*=9	*m*=10
/y/	glide, V	1.2958	1.3744	1.445
/w/	glide, V	0.301601	0.322598	0.343601
/l/	liquid, V	0.621002	0.659401	0.694599
/r/	liquid, V	0.916199	0.981403	1.0466
/m/	nasal, V	2.2416	2.4318	2.6132
/n/	nasal, V	2.4948	2.6632	2.8242
/f/	fricative, U	5.9298	6.4766	7.03994
/TH/	fricative, U	7.53824	8.1305	8.69302
/s/	fricative, U	7.41566	8.33818	9.60784
/SH/	fricative, U	6.6628	7.43038	8.15144
/h/	fricative, U	2.1832	2.2432	2.2856
/DH/	fricative, V	4.1374	4.5968	5.0402
/z/	fricatives, V	2.241	2.4026	2.554
/v/	fricative, V	1.3374	1.4426	1.5436

Pho-nem	Type, Voiced/ Unvoiced	*m*=8	*m*=9	*m*=10
/t/	stop, U	8.5708	10.2586	7.40088
/k/	stop, U	3.59	3.7522	3.8922
/p/	stop, U	5.5608	5.99326	6.4162
/d/	stop, V	3.97914	4.32532	4.64544
/g/	stop, V	1.278	1.3256	1.3564
/b/	stop, V	3.13856	3.40088	3.64058
/j/	affricative, V	2.809	2.966	3.1156
/CH/	affricative, U	2.3044	2.379	2.4364

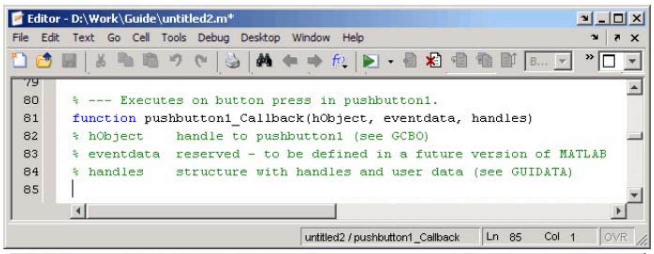
III. Laboratory Software Application: Speech Recognition

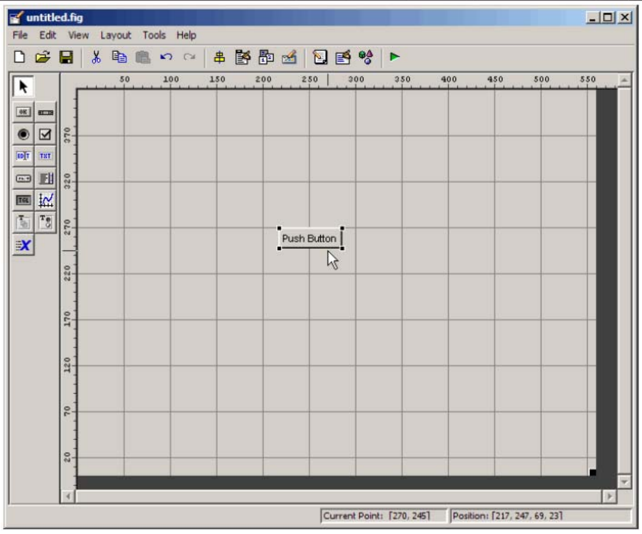
A. *Software Description*

Matlab® is the main software tool used for the laboratory by students. Matlab integrates computation, visualization, and programming in an easy-to-use environment where problems and solutions are expressed in familiar mathematical notation. The basic data element is an array that does not require dimensioning; this allows the user to solve many technical computing problems with matrix and vector formulations.

Beyond other features, this software allows both "programming in the small" to rapidly create quick and dirty

throw-away programs, and "programming in the large" to create large and complex application programs. Attending to the necessity of creating complex programs to be used as computer applications, Matlab offers an easy-to-use graphical interface called GUIDE. All the algorithms and code generated for our laboratory use GUIDE to present all the output information in an easy and interactive way. Some elements of GUIDE programming interface can be seen in fig. 12 and 13.

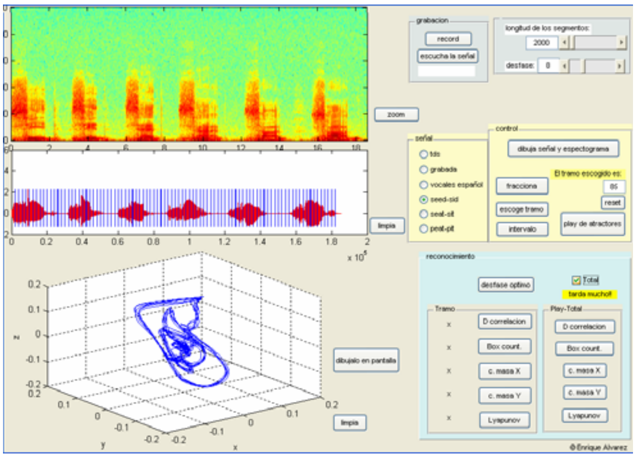




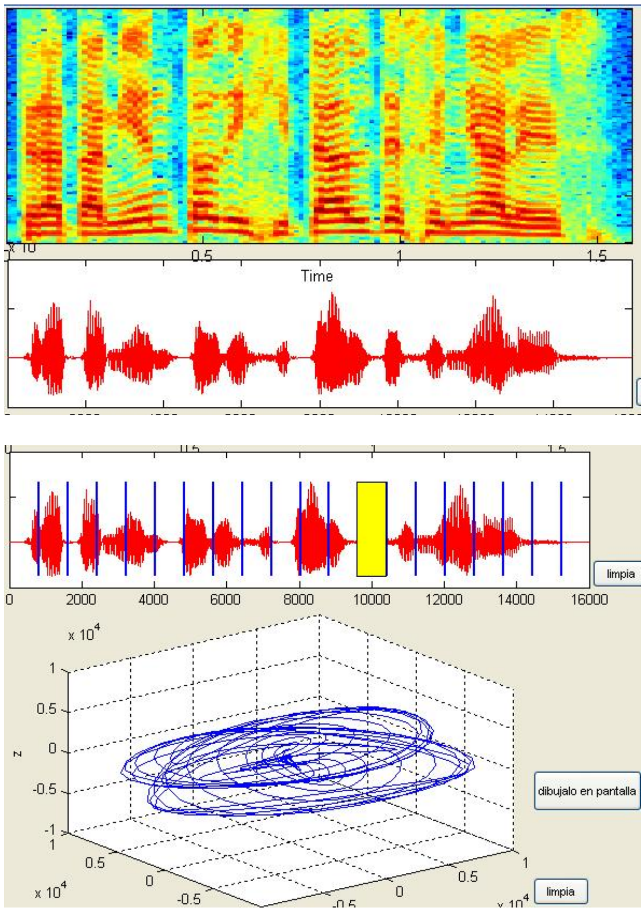
For further complete information about this essential engineering tool, the autor is remitted to Matlab manuals or books available in the technical literature.

B. Software Application Description

The main Matlab application defined for the laboratory contains, basically, almost all the graphic interfaces of the program. In this main window (figure 22), the user could either select one of the available signals or acquire a particular one (sound registration with a 10000Hz sampling frequency) in order to apply the different algorithms programmed and available to be used.



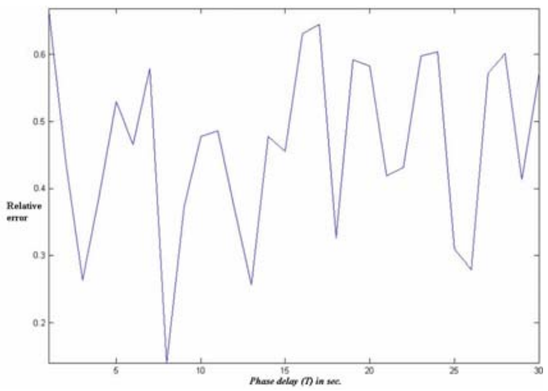
It is possible to change the way results are plotted. The main analysis includes signal spectrogram, sampled signal and the 3D attractor for a particular portion of signal selected from the sampled signal plot, just with two mouse clicks. These are the basis for the complete desired non-linear signal analysis: there are buttons to carry out the different algorithms programmed. Some details are shown in figure 23



Notice that all the software tool headings are in spanish, as the software is going to be used by spanish-speaker users.

For the first two laboratory sessions, students are given some algorithms for some of these analysis parameters. They

are told to program and insert them into the main application that they are given. Among others, they have to deal with algorithms related to the obtention of the proper embedding dimension *m*, modification of the proposed correlation dimension algorithm to reduce computational costs (N^2 to $N\log(N)$). At last, the main non-linear parameters of the signal are obtained; as an example, figure 15 plots the result for the obtention of the proper delay *T*.



Calculo del error debido a T

IV. Conclusion

This work presents a complete software tool, with graphic interface, as the learning basis for the ''Biomedical Signals and Algorithms' laboratory, which is an optional subject of the Telecommunication Degree (Technical University of Madrid, Spain). This provided software tool developes a significant part of attractor theory applied to speech recognition. As starting point, for the student, this non-linear view constitutes a new alternative to common frequency domain approach.

Considering the friendly graphic environment given by Matlab, the student will modify or create new algorithms for the obtention of a variety of non-linear parameters. As final evaluatory work for this laboratory, students are demanded to search an specific topic (scientific books or papers) related to bioengineering and to create a similar application, including all the algorithms needed to this purpose.

Acknowledgment

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